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PTO/SB/05 (08-00) (modified)
Approved for use through 9/30/2001, OMB 0651-0032
Patent and Trademark Office: U.S. DEPARTMENT OF COMMERCE

jc961 U.S. PTO 10/27/00

UTILITY PATENT APPLICATION TRANSMITTAL

(only for new nonprovisional applications under 37 CFR 1.53(b))

Attorney Docket Number	21706-05327
First Named Inventor	James H. Parry
Title	DISTORTION COMPENSATION IN AN ACOUSTIC ECHO CANCELER
Express Mail Label No.	EL541495171US

jc922 U.S. PTO 09/699188 10/27/00

APPLICATION ELEMENTS

1. ☒ Fee Transmittal Form (in duplicate)
2. ☐ Applicant claims small entity status. See 37 CFR 1.27
3. ☒ Specification — *Total Pages* 24
   (preferred arrangement set forth below)
   - Descriptive Title of the Invention
   - Cross Reference(s) to Related Case(s)
   - Statement Regarding Fed sponsored R & D
   - Background of the Invention
   - Brief Summary of the Invention
   - Brief Description of the Drawing(s)
   - Detailed Description
   - Claim or Claims
   - Abstract of the Disclosure
4. ☒ Drawing(s) *(35 U.S.C. 113)* — *Total Sheets* 3
5. Oath or Declaration
   a. ☒ New Declaration — *Total Pages* 2
      ☒ Executed (original or copy)
   b. ☐ Copy from a prior application (37 CFR 1.63(d))
      (for continuation/divisional with Box 17 completed)
      i. ☐ DELETION OF INVENTOR(S)
         Signed statement attached deleting inventor(s) named in the prior application, see 37 CFR 1.63(d)(2) and 1.33(b).
6. ☐ Application Data Sheet. See 37 CFR 1.76

ACCOMPANYING APPLICATION PARTS

7. ☒ Assignment Papers (cover sheet & document(s))
8. ☐ Certified Copy of Priority Document(s) *(if foreign priority is claimed)*
9. ☐ Information Disclosure Statement & PTO-1449
   ☐ Copies of IDS Citation(s)
10. ☐ Preliminary Amendment
11. ☒ Return Postcard
12. ☐ ____
13. ☐ ____
14. ☐ ____
15. ☐ ____
16. ☐ ____

ADDRESS TO:

Box Patent Application
Commissioner for Patents
Washington, D.C. 20231

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17. **If a CONTINUING APPLICATION,** *check appropriate box and supply the requisite information below and in a preliminary amendment or in an Application Data Sheet under 37 CFR 1.76:*
☐ Continuation ☐ Divisional ☐ Continuation-in-part (CIP) of prior application No: ___/_______
Prior application information: *Examiner:* ____________ *Group/Art Unit:* ________

For CONTINUATION OR DIVISIONAL APPS only: The entire disclosure of the prior application, from which an oath or declaration is supplied under Box 5b, is considered a part of the disclosure of the accompanying continuing or divisional application and is hereby incorporated by reference. The incorporation can only be relied upon when a portion has been inadvertently omitted from the submitted application parts

18. CORRESPONDENCE ADDRESS

☐ *Customer Number or Bar Code Label* ***(Insert Customer No. or Attach bar code label here)*** *Or* ☒ *Correspondence address below*

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Signature	[signature]	*Date*	October 27, 2000

21706/05327/SF/5034497.1

EXPRESS MAIL NO. EL541495171US

PTO/SB/17 (10-00)(modified)
Approved for use through 09/29/01, OMB 0651-0032
Patent and Trademark Office: U.S. DEPARTMENT OF COMMERCE

0002/PTO(modified) U.S. Department of Commerce
Rev. 10/2000 Patent and Trademark Office

FEE TRANSMITTAL

TOTAL AMOUNT OF PAYMENT

Subtotal (1) + Subtotal (2) + Subtotal (3) = **($)1,308.00**

Complete if Known	
Application Number	NEW
Filing Date	HEREWITH
First Named Inventor	James H. Parry
Group Art Unit	UNASSIGNED
Examiner Name	UNASSIGNED
Attorney Docket Number	21706-05327

METHOD OF PAYMENT

1. The Commissioner is hereby authorized to:

☐ Charge the indicated fees to the below mentioned deposit account.

☒ Charge any additional fee required under 37 CFR 1.16 - 1.21 or credit any over payments to the below mentioned deposit account. †

☐ Applicant claims small entity status See 37 CFR 1 27

Deposit Account Number: 19-2555
***Deposit Account Name:* FENWICK & WEST LLP**

A Duplicate Copy of this authorization is attached

2. ☒ Payment Enclosed:
☒ Check ☐ Credit Card ☐ Other

FEE CALCULATION (fees effective 10/01/2000)

1. FILING FEE

Large Entity Fee Code/Fee	Small Entity Fee Code/Fee	Fee Description	Fee Due
101/$710	201/$355	Utility Filing	710
106/$320	206/$160	Design Filing	
108/$710	208/$355	Reissue	
114/$150	214/$75	Provisional Filing	

SUBTOTAL (1) ($) 710

2. CLAIMS

Large Entity Fee Code/Fee	Small Entity Fee Code/Fee	Fee Description
103/$18	203/$9	Claims in excess of 20
102/$80	202/$40	Independent claims in excess of 3
104/$270	204/$135	Multiple dependent claim
109/$80	209/$40	Reissue independent claims over original patent
110/$18	210/$9	Reissue claims in excess of 20 and over original patent

FEE CALCULATION (continued)

3. ADDITIONAL FEES

Large Entity Fee Code/Fee	Small Entity Fee Code/Fee	Fee Description	Fee Due
105/$130	205/$65	Surcharge - late filing fee or oath	
127/$50	227/$25	Surcharge-late provisional filing fee or cover sheet	
147/$2,520	147/$2,520	For filing a request for reexamination	
115/$110	215/$55	Extension for response within first month†	
116/$390	216/$195	Extension for response within second month†	
117/$890	217/$445	Extension for response within third month†	
118/$1,390	218/$695	Extension for response within fourth month†	
128/$1,890	228/$945	Extension for response within fifth month†	
119/$310	219/$155	Notice of Appeal	
141/$1,240	241/$620	Petition to revive unintentionally abandoned application	
142/$1,240	242/$620	Utility Issue Fee (Or Reissue)	
143/$440	243/$220	Design Issue Fee	
122/$130	122/$130	Petitions to the Commissioner	
126/$240	126/$240	Submission of Information Disclosure Statement	
179/$710	279/$355	Request for Continued Examination (RCE)	
581/$40	581/$40	Recording each patent assignment per property (times number of properties)	40
146/$710	246/$355	Filing a submission after final rejection (37 CFR 1.129(a))	
149/$710	249/$355	For each additional invention to be examined (37 CFR 1.129(b))	
	Other fee *(specify)*:		
	Other fee *(specify)*:		

SUBTOTAL (3) ($) 40

For	(Col 1) No. of Existing Claims		(Col. 2) Highest No Previously Paid For		(Col. 3) Extra**		Fee		Fee Due
TOTAL	51	minus*	20 or 0	=	31	x	18	=	558
INDEP	3	minus*	3 or 0	=	0	x	0	=	0
[ ] First presentation of multiple dependent claim								=	

* *Subtract the greater number of Col. 2*
** *If the difference between Col. 1 and Col. 2 is less than zero, then enter "0" in Col. 3*

SUBTOTAL (2) ($) 558

SUBMITTED BY		Complete (if applicable)	
Typed or Printed Name	**Brian M. Hoffman**	Reg Number	**39,713**
Signature	[signature]	Date	**October 27, 2000**

† *Request for Extension of Time per 37 CFR 1.136 (a)(3) made hereby*
Rev. 09/30/2000

21706/05327/SF/5034390.1

EXPRESS MAIL NO. EL541495171US

DISTORTION COMPENSATION IN AN ACOUSTIC ECHO CANCELER

INVENTOR

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EXPRESS MAIL NO. EL541495171US

DISTORTION COMPENSATION IN AN ACOUSTIC ECHO CANCELER

INVENTOR

JAMES H. PARRY

BACKGROUND

FIELD OF THE INVENTION

This invention pertains in general to telephony and televideo conferencing and in particular to performing acoustic echo cancellation on potentially distorted audio signals.

BACKGROUND ART

Two-way audio communications systems, such as speakerphones and video communications systems having audio capabilities, utilize both a microphone and a loudspeaker. The microphone transmits speech and other sounds from the local terminal to remote terminals while the loudspeaker emits sounds received from remote terminals. In a typical hands-free system, the loudspeaker and microphone are located in close proximity and sounds produced by the loudspeaker are picked up by the microphone. Without signal processing, therefore, a feedback loop is easily created between the loudspeaker and microphone. This feedback can cause the loudspeaker to emit an undesirable "howling" noise and cause the remote terminals to hear echoes.

One simple technique for eliminating feedback is to provide half-duplex switching where only the microphone or the loudspeaker is active at any given instant. In a typical half-duplex system, the loudspeaker is active until a sound is detected at the microphone. Then, the loudspeaker becomes inactive and the microphone becomes active for the

duration of the sound. Half-duplex systems have many inherent problems, not the least of which is that a slight noise may unintentionally cause the loudspeaker to cut out. As a result, it is often difficult to conduct a normal conversation with a system using half-duplex switching.

More sophisticated audio communications systems use acoustic echo cancellation (AEC) to reduce echoes and eliminate howling. An AEC system typically utilizes a sample-by-sample copy of the signal going to the loudspeaker as the basis for an estimate of the echo returning through the microphone, as taught in U.S. Patent No. 4,965,822, entitled FULL DUPLEX SPEAKERPHONE, which issued on October 23, 1990 and is incorporated by reference herein. This estimated echo is subtracted on a sample-by-sample basis in an attempt to separate out only that portion of the microphone signal due to sounds coming from sources other than the speaker. An adaptive AEC uses a filter having slowly adjusted weights to form the echo estimate in an effort to more accurately subtract the echo from the returned audio signal. Subsequent conditioning performed on the output of the AEC may include automatic gain control (AGC) and perceived noise reduction.

A problem with the above approach is that the loudspeakers do not produce sound pressure signals that are exactly proportional to the driving voltage (or current). Likewise, microphones are imperfect in an analogous sense. There may also be other sources of distortion within the sound system, such as amplifiers, analog-to-digital (A/D) and digital-to-analog (D/A) converters, and perhaps even the user's environment. Existing AEC systems do not accurately remove the nonlinear components of the returned signal due to these sources of potential distortion. As a result, a badly distorted form of

the echo can pass through the echo cancellation process. Another undesirable effect of these introduced distortions is that the adaptation of the AEC parameters is degraded, leading to a greater perceived echo.

One potential solution to the problem of degraded AEC adaptation is to use a reduced adaptation rate during periods of very loud sound output. This technique is used, for example, in U.S. Application No. 09/XXX,XXX, entitled APPARATUS AND METHOD FOR CONTROLLING AN ACOUSTIC ECHO CANCELER, filed on January 20, 2000, and incorporated by reference herein. However, reducing the adaptation rate has the undesirable effect of slowing the system's response to a changing acoustic environment such as when users are in motion and/or the room temperature fluctuates.

Another potential solution is to use higher quality loudspeakers and other components. This solution, however, carries with it considerable expense and places severe limitations on the designs of the equipment. High-quality loudspeakers are typically large and heavy and generate strong external magnetic fields. Often, the audio communications system is integrated into another sound system, such as the audio subsystem of a laptop computer, where a high-quality loudspeaker cannot be used.

Therefore, there is a need for a technique for more accurately estimating the echo when performing acoustic echo cancellation. There is also a need for a technique for more accurately adapting the estimated echo in response to changing acoustic characteristics.

DISCLOSURE OF THE INVENTION

The above needs are met by using modules to estimate the nonlinear distortions in the audio signal returned from the microphone that were introduced by the loudspeaker, microphone, and related components.

A typical audio communications system has a plurality of terminals coupled to a switch. The terminals can include, for example, dedicated speakerphones, desktop handsets with or without speakerphone capabilities, cellular phones, and/or personal computer (PC) systems with audio capabilities. The switch may be dedicated to audio communications, as is a private branch exchange (PBX), or distributed and multifunctional, as is an Internet server.

Each terminal preferably includes a microphone and a loudspeaker. An amplifier amplifies the electrical signals produced by the microphone and provides its output to an A/D converter. The A/D converter outputs equivalent digital samples. The loudspeaker is driven by another amplifier which, in turn, is driven by the output of a D/A converter. The D/A converter receives digital samples representing the sound pressure waves to be produced by the loudspeaker.

In order to cancel echoes of the loudspeaker picked up by the microphone, the audio communications system has an acoustic echo cancellation (AEC) module. The AEC module can be located in the terminal or elsewhere in the audio communications system. U.S. Patent Application No. 09/660,205, entitled COMMUNICATIONS SYSTEM AND METHOD UTILIZING CENTRALIZED SIGNAL PROCESSING, filed on September 12, 2000, and incorporated by reference herein, describes potential

locations of the AEC module. The AEC module preferably receives the digital signal sent to the loudspeaker and the digital signal received from the microphone.

The digital loudspeaker signal is processed by an audio generation module (AGM) to model the substantially nonlinear distortions that can occur during the process of playing the audio signal at the loudspeaker. The AGM includes a modeling path comprised of one or more distortion modules. Each distortion module receives digital samples as input, modifies the samples to model a form of distortion, and outputs the modified samples. A distortion module can be adaptive or it can be partly or wholly pre-established. Preferably, the AGM can add or remove distortion modules from the modeling path at any time in response to characteristics of the digital samples or under direction from other modules. Distortions that can be modeled by the distortion modules in the AGM modeling path include, for example, amplifier clipping, loudspeaker voice coil displacement, harmonic distortion introduced by the loudspeaker, and hysterisis in an iron-core inductor.

The AGM outputs digital sample values to an acoustic echo estimation (AEE) module. The AEE module preferably uses known adaptive algorithms to adapt the digital samples to compensate for substantially linear changes in the echo characteristics of the environment in which the loudspeaker and microphone are located. For example, the AEE module can modify the digital samples to account for changes in echo attenuation due to relocation of people in the vicinity of the microphone.

The output of the AEE module is received by an audio sensing module (ASM). The ASM performs a function similar to the AGM, except that the ASM models substantially nonlinear distortions that occur while sensing the audio signal. Accordingly,

the ASM models distortions such as microphone 116 centerclipping, amplifier zero crossing distortion, saturation in either the microphone or the amplifier, and distortions introduced by the A/D converter. The output of the ASM represents the estimate of the echo of the loudspeaker signal in the signal received from the microphone.

The digital samples returned from the microphone and the output of the ASM are received by an adder module. The adder module subtracts the estimated echo received from the ASM from the samples returned from the microphone, thereby removing at least part of the estimated echo of the loudspeaker from the microphone signal.

BRIEF DESCRIPTION OF THE DRAWINGS

FIGURE 1 is a high-level block diagram of an audio communications system according to an embodiment of the present invention;

FIGURE 2 is a block diagram illustrating various components of the audio communications system including an acoustic echo cancellation (AEC) module; and

FIGURE 3 is a lower-level view of an exemplary audio generation module in the AEC module.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

FIG. 1 is a high-level block diagram of an audio communications system 100 according to an embodiment of the present invention. A plurality of terminals 110A-D are coupled to a switch 112 via communications links 114A-D. The terminal types can be heterogeneous or homogeneous. In one embodiment, the terminals include: dedicated speakerphones, desktop handsets with or without speakerphone capabilities, cellular

phones, and/or personal computer (PC) systems with audio capabilities. As used herein, the phrase "audio communications system" also includes video conferencing systems having audio capabilities. Each terminal 110, of which terminal 110A is representative, preferably includes a microphone 116A and a loudspeaker 118A. As is known in the art, the microphone 116 converts sound pressure waves into electrical signals and the loudspeaker 118 converts electrical signals into sound pressure waves.

The communications links 114 carry audio data representative of sounds picked up by the microphone 116 and to be played by the loudspeaker 118 to/from the switch 112. The communications links 114 may be wired or wireless. Moreover, the links 114 may include dedicated private links, shared links utilizing a publicly-accessible telephone network, and/or links using a public or private data communications network such as the Internet. Data traveling over the links 114 may pass through one or more switches or link types before reaching the switch 112 or terminal 110, although a preferred embodiment of the present invention treats a link passing through multiple links and switches as a single logical link. The data carried by the communications links 114 can be digital and/or analog. If the data is digital, it is preferably transmitted as a series of discrete data packets, such as Internet protocol (IP) packets. In one embodiment, the digital data is encoded into a compressed format.

The switch 112 switches and routes communications among the terminals. The switch 112 can be, for example, a private branch exchange (PBX) located at a business or other entity, a publicly-accessible switch operated by a telephone company or other entity providing audio communications, or an Internet server supporting Internet telephony.

Thus, the term "switch" includes any device or combination of devices capable of providing the switching and other functionality attributed to the switch herein.

In one embodiment, the terminals 110 and/or switch 112 have one or more of the components found in a typical computer system, including a processing unit, random access memory (RAM), read-only memory (ROM), a storage device such as a hard drive, and/or other hardware and software for providing the functionality described herein. Aggregations of machine-executable code, data, circuitry, and/or data storage areas for performing a specific purpose or purposes are referred to as "modules." Different modules may share common code, data, and/or circuitry. The modules include, for example, signal processing modules, digital-to-analog (D/A) and analog-to-digital (A/D) converter modules, and amplifier modules. Modules may hold in their storage areas previous values of signals and current statistics derived therefrom. Modules can also use adaptive techniques, or training, to perform the modules' functionalities. As used herein, the terms "adaptation" and "training" are interchangeable and refer to acting on a signal responsive to previous values of that signal or other signals, statistics derived from the signals, and/or external controls or sensors.

FIG. 2 is a block diagram illustrating various components of the audio communications system including an acoustic echo cancellation (AEC) module 210. FIG. 2 illustrates the microphone 116 of FIG. 1 having its output coupled to an amplifier 212. As is known in the art, the microphone 116 converts sound pressure waves into electrical signals. The amplifier 212 amplifies the electrical signals and provides its output to an A/D converter 214. The A/D converter 214 outputs digital sample values representative of the sound pressure waves to the AEC module 210.

FIG. 2 also illustrates the speaker 118 of FIG. 1. The speaker 118 generates sound pressure waves in response to an input received from an amplifier 216. The amplifier 216, in turn, is driven by the output of a D/A converter 218. The D/A converter 218 receives digital sample values representing the sound pressure waves as input from the link 114 or another source.

In general, the AEC module 210 estimates the echo of sounds played by the loudspeaker 118 that are picked up by the microphone 116, subtracts the estimated echo from the microphone's audio signal, and outputs the resulting echo-cancelled signal. In one embodiment, the AEC module 210 is located in the terminal 110. Accordingly, the output of the AEC module 210 is passed over the communications links 114 to the switch 112. In alternative embodiments, the AEC module 210 is located in the switch 112 or anywhere else that echo cancellation is desired and representations of the loudspeaker and microphone signals are available.

Turning to the AEC module 210 itself, the digital samples representing the audio signal sensed by the microphone output by the A/D converter 214 are received by an adder module 220. The adder module 220 also receives an input 222 providing digital samples representing the echo from the loudspeaker 118 estimated to be present in the microphone signal. The adder module 220 preferably adds the negative of the estimated echo to the signal received from the A/D converter 214. Preferably, the adder module 220 works on a sample-by-sample basis. In one embodiment, both the estimated echo samples received from the input 222 and the sample values received from the A/D converter 214 bear sequencing information that the adder module 220 uses to match the

samples. U.S. Patent Application No. 09/660,205, incorporated by reference herein, discloses additional details related to the sequencing information.

The output of the adder module 220 is passed to a perceived noise reduction module 224. This module 224 preferable reduces perceived noise in the audio signal. Techniques for reducing perceived noise are well known in the art.

The output of the perceived noise reduction module is preferably passed to an automatic gain control (AGC) module 226. As is known in the art, the AGC module 226 preferably isolates times during which local speech is thought to be present in the input signal and adjusts the signal gain so that the speech is near a predetermined level when considered on average. The AGC module 226 can use adaptive techniques to perform AGC. The output 228 of the AGC module 226 is preferably provided to the switch 112 via the communications links 114 as described above.

The AEC module 210 also receives an input 230 carrying digital sample values representing the audio signal being sent to the loudspeaker 118 of the terminal 110. If the AEC module 210 is located in the terminal 110, then this input 230 is received from the switch 112 via the communications links 114. The loudspeaker 118 digital sample values are received by an audio generation module (AGM) 232 within the AEC module 210.

The AGM 232 preferably modifies the digital sample values to model substantially nonlinear distortions that can occur during the process of generating the audio signal. FIG. 3 is a block diagram illustrating a lower-level view of the AGM 232 according to an exemplary embodiment of the present invention. The AGM 232 includes a modeling path 310 comprised of logical interconnects among one or more distortion modules 312 that operate on the digital samples traveling through the path. Each

distortion module 312 receives digital samples as input, modifies the samples to model a form of distortion, and outputs the modified samples. Preferably, the AGM 232 can add or remove distortion modules from the modeling path 310 at any time in response to characteristics of the digital samples or under direction from other modules.

The AGM 232 preferably models effects which are substantially nonlinear. Certain embodiments utilize artificial neural networks (ANNs) to achieve adaptation. Those ANNs which are not adaptive may be present at the time of manufacture and do not require feedback for further adaptation. ANNs in adaptive modules 312 utilize internal and/or external feedback. Such feedback may be from other distortion modules 312, from the loudspeaker digital signal, and/or from the microphone signal before or after the adder module 220. These many possible feedback paths have been omitted from the modeling path 310 in FIG. 3 in order to clarify the teachings of the present invention.

The example of a modeling path 310 illustrated in the AGM 232 of FIG. 3 has three distortion modules 312A, 312B, 312C arranged in sequence. Each distortion module 312 preferably contains a filter or other operator that acts on the input samples. The module 312 can be adaptive or it can be partly or wholly pre-established. Likewise, the module 312 can operate in the time or frequency domains. The module 312 can also act in response to short or long-term signal characteristics to model effects such as heat build-up.

Preferably, each distortion module 312 independently models a form of distortion. In FIG. 3, the first distortion module 312A models amplifier clipping by enforcing a hard limit on signal amplitudes. Thus, this distortion module 312A models the effects of the speaker amplifier 216 in the terminal 110 on the analog signal sent to the loudspeaker

118. The second distortion module 312B models loudspeaker 118 voice coil (or equivalent structure) displacement. In one embodiment, this distortion module 312B estimates the nonlinear relationship between the voice coil displacement and the driving current. In one embodiment, the driving current estimate received by the voice coil displacement module 312B is generated by the amplifier clipping module 312A and so may be a nonlinear representation of the loudspeaker digital samples. The third distortion module 312C models harmonic distortion introduced by the loudspeaker 118. In one embodiment, this distortion module 312C applies harmonic distortion with a strength modulated by the energy in the spectral components subject to the distortion. Thus, this distortion module 312C mimics the operation of a loudspeaker 118 driven with high electrical amplitudes where diaphragms distort and resonate. Other distortion modules 312 that may be utilized in the modeling path 310 include modules that account for distortions introduced by the D/A conversion module 218 and modules that account for hysterisis in iron core inductors.

In one embodiment of the present invention, the distortion modules 312 are tailored to model the distortions introduced by specific types of hardware. For example, if the AGM 232 is located in the terminal 110, the amplifier clipping 310A and voice coil displacement 310B modules can be specifically tailored for the amplifiers and voice coils included in the terminal 110.

The AGM 232 outputs digital sample values representing the distorted audio signal to an acoustic echo estimation (AEE) module 234. The AEE module 234 preferably uses adaptive algorithms to adapt the digital samples to compensate for substantially linear changes in the echo characteristics of the environment in which the

loudspeaker 118 and microphone 116 are located. For example, the AEE module 234 can modify the digital samples to account for changes in echo attenuation due to relocation of people in the vicinity of the microphone 116.

The digital sample values output by the AEE module 234 are preferably received by an audio sensing module (ASM) 236. The ASM 236 preferably modifies the digital sample values to model distortions that can occur in the process of sensing the audio signal. Like the AGM 232, the ASM 236 preferably includes a modeling path comprised of logical interconnects among one or more distortion modules. The modeling path for the ASM 236 is not shown in the figures because it would be redundant in view of FIG. 3. Also like the AGM 232, the ASM 236 preferably models substantially nonlinear distortions. Unlike the AGM 232, the ASM preferably models distortions such as microphone 116 centerclipping, amplifier zero crossing distortion, saturation in either the microphone or the amplifier, and/or distortions introduced by the A/D converter 214. The output of the ASM 236 is provided to the adder module 220 and becomes the input signal representing the echo from the loudspeaker 118 estimated to be present in the microphone signal described above.

Accordingly, the AEC module 210 of the present invention accurately models the effects of distortion on the audio signals. The modeled types of distortion include nonlinear distortions introduced while generating and sensing the audio signal and linear echoes introduced responsive to room characteristics. This distortion modeling enables the AEC to more accurately cancel the echo in the signal received from the microphone 116.

The above description is included to illustrate the operation of the preferred embodiments and is not meant to limit the scope of the invention. The scope of the invention is to be limited only by the following claims. From the above discussion, many variations will be apparent to one skilled in the relevant art that would yet be encompassed by the spirit and scope of the invention.

CLAIMS

I claim:

1. A module in an audio communications system, comprising:
a first input for receiving a first audio signal;
a second input for receiving a second audio signal, wherein at least a portion of the second audio signal is an echo of the first audio signal;
a distortion module receiving the first audio signal, the distortion module adapted to model a distortion on the first audio signal and produce a distorted signal; and
an adder module for receiving the distorted signal and the second audio signal and adapted to use the distorted signal to remove at least part of the echo from the second audio signal.

2. The module of claim 1, wherein the first and second audio signals bear sequencing information and wherein the adder module is adapted to use the sequencing information to remove at least part of the echo from the second audio signal.

3. The module of claim 1, further comprising:
an audio generation module receiving the first audio signal and adapted to use the distortion module to model a distortion that occurs responsive to playing the first audio signal through a loudspeaker.

4. The module of claim 3, wherein the audio generation module comprises:
a modeling path having one or more distortion modules that model distortions on the first audio signal.

5. The module of claim 4, wherein each distortion module models a different type of distortion.

6. The module of claim 4, wherein the audio generation module alters the modeling path in real-time responsive to distortions that may occur on the first audio signal.

7. The module of claim 3, wherein the distortion module models an effect of amplifier clipping on the first audio signal.

8. The module of claim 3, wherein the distortion module models an effect of voice coil displacement on sound pressure waves produced by the loudspeaker responsive to the first audio signal.

9. The module of claim 3, wherein the distortion module models an effect of hysterisis in an iron core inductor on the first audio signal.

10. The module of claim 3, wherein the distortion module models an effect of harmonic distortion on sound pressure waves produced by the loudspeaker responsive to the first audio signal.

11. The module of claim 1, further comprising:

an acoustic echo estimation module receiving the first audio signal and for adapting the first audio signal to compensate for substantially linear changes in the second audio signal.

12. The module of claim 1, further comprising:

an audio sensing module receiving the first audio signal and adapted to use the distortion module to model a distortion that occurs responsive to sensing the second audio signal.

13. The module of claim 12, wherein the audio sensing module comprises:

a modeling path having one or more distortion modules that model distortions on the second audio signal.

14. The module of claim 13, wherein each distortion module models a different type of distortion.

15. The module of claim 13, wherein the audio sensing module alters the modeling path in real-time responsive to distortions that may occur on the second audio signal.

16. The module of claim 12, wherein the distortion module models an effect of microphone centerclipping on the second audio signal.

17. The module of claim 12, wherein the distortion module models an effect of amplifier zero crossing distortion on the second audio signal.

18. The module of claim 1, wherein the distortion module models a pre-established distortion.

19. The module of claim 1, wherein the distortion module is adaptive.

20. The module of claim 1, wherein the distortion module models a nonlinear distortion.

21. The module of claim 1, wherein the distortion module operates in a frequency domain.

22. A method of canceling an echo in an audio signal, comprising the steps of:

receiving a first audio signal;

receiving a second audio signal, wherein at least a portion of the second audio signal is a distorted echo of the first audio signal;

modeling one or more types of distortions on the first audio signal to produce a distorted audio signal; and

subtracting the distorted audio signal from the second audio signal to at least partially cancel the distorted echo of the first audio signal from the second audio signal.

23. The method of claim 22, wherein the modeling step comprises the step of: adaptively modeling one or more types of distortion.

24. The method of claim 22, wherein the modeling step comprises the step of: modeling a pre-established type of distortion.

25. The method of claim 22, further comprising the step of: retrieving sequencing information from the first and second audio signals; wherein the subtracting step uses the sequencing information to at least partially cancel the distorted echo of the first audio signal from the second audio signal.

26. The method of claim 22, wherein the modeling step comprises the step of: passing the first audio signal through a modeling path comprising one or more distortion modules, each distortion module applying a type of distortion to the first audio signal.

27. The method of claim 26, wherein the modeling path models distortions that occur responsive to playing the first audio signal through a loudspeaker.

28. The method of claim 27, wherein the passing step comprises the step of: passing the first audio signal through a distortion module that models an effect of amplifier clipping on the first audio signal.

29. The method of claim 27, wherein the passing step comprises the step of: passing the first audio signal through a distortion module that models an effect of voice coil displacement on sound pressure waves produced by the loudspeaker responsive to the first audio signal.

30. The method of claim 27, wherein the passing step comprises the step of: passing the first audio signal through a distortion module that models an effect of harmonic distortion on the sound pressure waves produced by the loudspeaker responsive to the first audio signal.

31. The method of claim 27, wherein the passing step comprises the step of: passing the first audio signal through a distortion module that models an effect of hysterisis in inductors containing iron on the first audio signal.

32. The method of claim 26, wherein the modeling path models distortions that occur responsive to sensing the second audio signal.

33. The method of claim 32, wherein the passing step comprises the step of: passing the first audio signal through a distortion module that models an effect of microphone centerclipping on the second audio signal.

34. The module of claim 32, wherein the passing step comprises the step of: passing the first audio signal through a distortion module that models an effect of amplifier zero crossing distortion on the second audio signal.

35. A terminal for an audio communications system, the terminal comprising: a loudspeaker for producing sound pressure waves responsive to a received first signal;

a microphone for converting sound pressure waves into a second signal, wherein a portion of the second signal represents an echo of the sound pressure waves produced by the loudspeaker;

a distortion module receiving the first signal and adapted to modify the first signal to model a type of distortion to produce a distorted signal; and

an adder module for removing at least a portion of the echo of the sound pressure waves produced by the loudspeaker from the second signal responsive to the distorted signal.

36. The terminal of claim 35, wherein the first and second signals bear sequencing information and wherein the echo cancellation module is adapted to use the sequencing information to remove at least part of the echo from the second signal.

37. The terminal of claim 35, further comprising:

an audio generation module receiving the first signal and adapted to use the distortion module to model a distortion that occurs responsive to playing the first signal through the loudspeaker.

38. The terminal of claim 37, wherein the audio generation module has a modeling path comprising one or more distortion modules that model distortions on the first signal.

39. The terminal of claim 38, wherein each of the one or more distortion modules models a different type of distortion.

40. The terminal of claim 38, wherein the audio generation module alters the modeling path in real-time responsive to distortions that may occur on the first signal.

41. The terminal of claim 37, wherein the distortion module models an effect of amplifier clipping on the first signal.

42. The terminal of claim 37, wherein the distortion module models an effect of voice coil displacement on the sound pressure waves produced by the loudspeaker.

43. The terminal of claim 37, wherein the distortion module models an effect of hysterisis in an iron core inductor on the first signal.

44. The terminal of claim 37, wherein the distortion module models an effect of harmonic distortion introduced by the loudspeaker on the sound pressure waves.

45. The terminal of claim 35, further comprising:

an acoustic echo estimation module receiving the first signal and for adapting the first signal to compensate for substantially linear changes in the second signal.

46. The terminal of claim 35, further comprising:

an audio sensing module receiving the first signal and adapted to use the distortion module to model a distortion that occurs responsive to sensing the audio signal.

47. The terminal of claim 46, wherein the audio sensing module has a modeling path comprising one or more distortion modules that model distortions on the second signal.

48. The terminal of claim 47, wherein each of the one or more distortion modules models a different type of distortion.

49. The terminal of claim 47, wherein the audio sensing module alters the modeling path in real-time responsive to distortions that may occur on the second audio signal.

50. The terminal of claim 46, wherein the distortion module models an effect of microphone centerclipping on the second signal.

51. The terminal of claim 46, wherein the distortion module models an effect of amplifier zero crossing distortion on the second signal.

DISTORTION COMPENSATION IN AN ACOUSTIC ECHO CANCELER

ABSTRACT OF THE DISCLOSURE

An audio communications system has an acoustic echo cancellation (AEC) module. The AEC module receives a digital signal sent to a loudspeaker and a digital signal received from a microphone. The signal received from the microphone contains an echo of the signal played through the loudspeaker. The loudspeaker signal is processed by an audio generation module (AGM) that models substantially nonlinear distortions that can occur while producing the signal played through the loudspeaker. The AGM includes a modeling path comprised of one or more distortion modules. Each distortion module receives digital samples as input, modifies the samples to model a form of distortion, and outputs the modified samples. The output of the AGM is provided to an acoustic echo estimation (AEE) module that uses adaptive algorithms to compensate for substantially linear changes in the echo characteristics of the environment in which the loudspeaker and microphone are located. The output of the AEE module is provided to an audio sensing module (ASM) that models substantially nonlinear distortions that can occur while receiving the signal from the microphone. The digital samples returned from the microphone, and the output of the ASM, are received by an adder module. The adder module subtracts the estimated echo from the samples returned from the microphone, thereby removing at least part of the estimated echo from the microphone signal.

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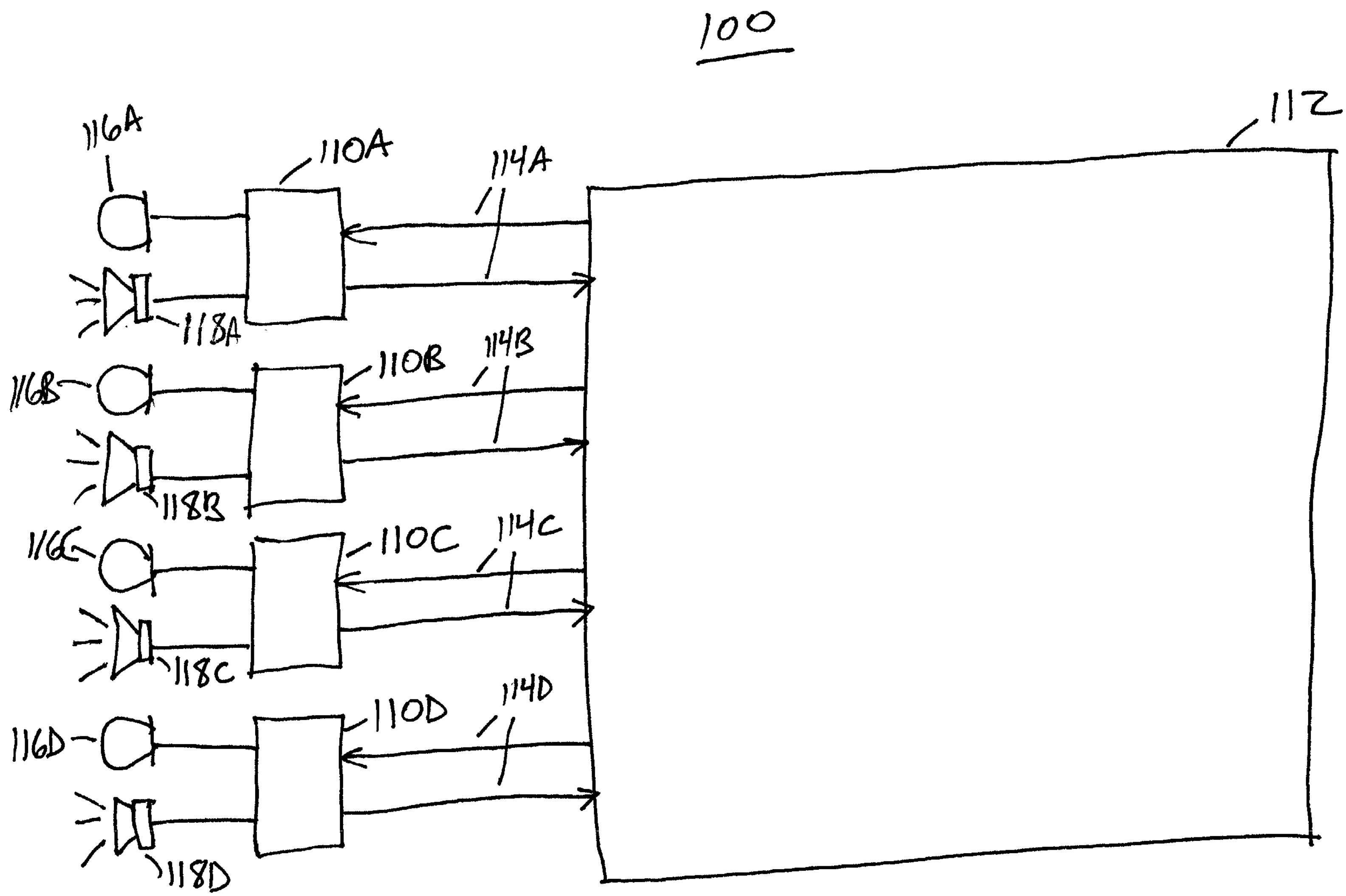
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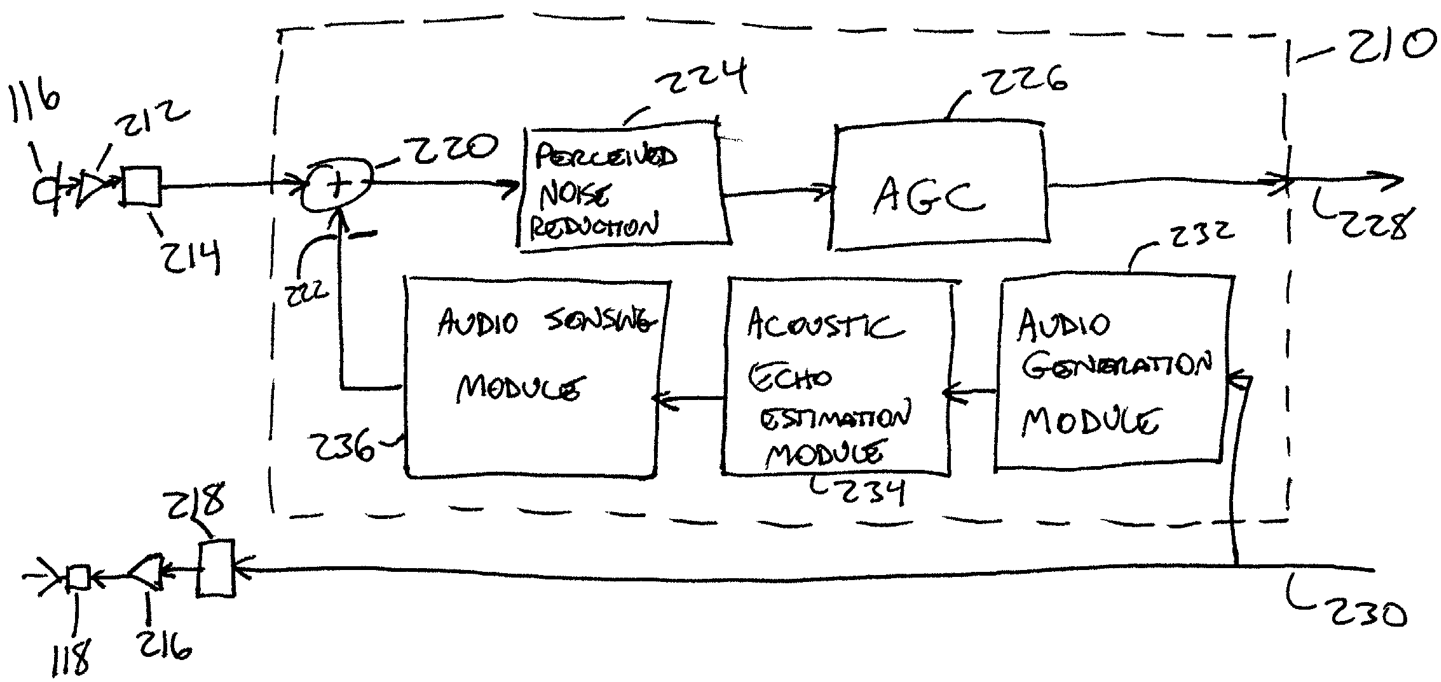
FIG. 1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212
214
220
222
224
PERCEIVED NOISE REDUCTION
226
AGC
228
210
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AUDIO GENERATION MODULE
ACOUSTIC ECHO ESTIMATION MODULE
234
AUDIO SENSING MODULE
236
218
216
118
230

FIG. 2

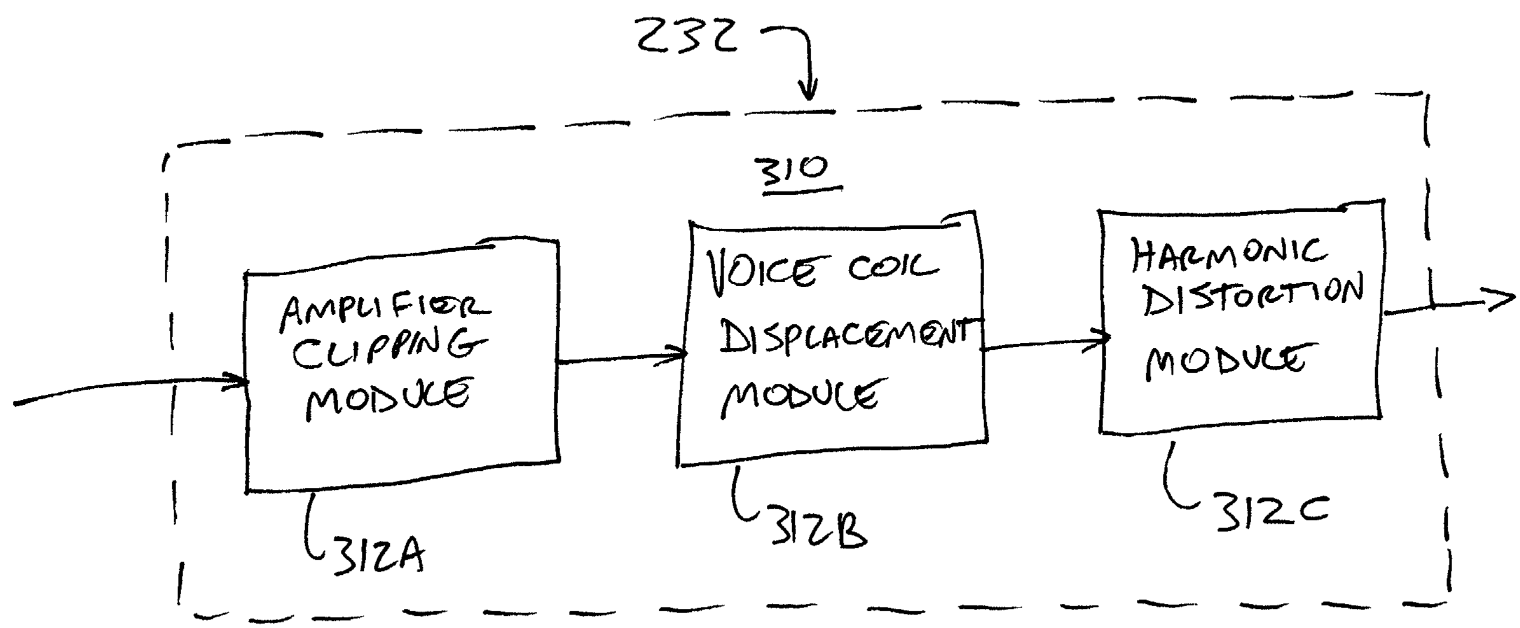
232
310
AMPLIFIER CLIPPING MODULE
VOICE COIL DISPLACEMENT MODULE
HARMONIC DISTORTION MODULE
312A
312B
312C

FIG. 3

Express Mail No. EL541495171US

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Patent and Trademark Office

DECLARATION FOR UTILITY OR DESIGN PATENT APPLICATION

[X] Declaration Submitted with Initial Filing OR [ ] Declaration Submitted after Initial Filing

Attorney Docket Number	21706-05327
First Named Inventor	James H. Parry
COMPLETE IF KNOWN	
Application Number	NEW
Filing Date	HEREWITH
Group Art Unit	UNASSIGNED
Examiner Name	UNASSIGNED

As a below named inventor, I hereby declare that:

My residence, post office address, and citizenship are as stated below next to my name.

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled:

DISTORTION COMPENSATION IN AN ACOUSTIC ECHO CANCELER

the specification of which (*Title of the Invention*)

[X] is attached hereto

OR

[ ] was filed on (MM/DD/YYYY) [____________] as United States Application Number or PCT International Application Number [____________] and was amended on (MM/DD/YYYY) [____________] (if applicable).

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims, as amended by any amendment specifically referred to above.

I acknowledge the duty to disclose information which is material to patentability as defined in Title 37 Code of Federal Regulations. § 1.56.

I hereby claim foreign priority benefits under Title 35, United States Code § 119 (a)-(d) or § 365(b) of any foreign application(s) for patent or inventor's certificate, or § 365 (a) of any PCT international application which designated at least one country other than the United States of America, listed below and have also identified below, by checking the box, any foreign application for patent or inventor's certificate, or of any PCT international application having a filing date before that of the application on which priority is claimed.

Prior Foreign Application Number(s)	Country	Foreign Filing Date (MM/DD/YYYY)	Priority Not Claimed	Certified Copy Attached? YES	NO
			[ ]	[ ]	[ ]
			[ ]	[ ]	[ ]
			[ ]	[ ]	[ ]
			[ ]	[ ]	[ ]
			[ ]	[ ]	[ ]

[ ] Additional foreign application numbers are listed on a supplemental priority sheet attached hereto:

I hereby claim the benefit under Title 35, United States Code § 119(e) of any United States provisional application(s) listed below.

Application Number(s)	Filing Date (MM/DD/YYYY)	[ ] Additional provisional application numbers are listed on a supplemental sheet attached hereto.

DECLARATION	Page 2

I hereby claim the benefit under Title 35, United States Code § 120 of any United States application(s), or § 365(c) of any PCT international application designating the United States of America, listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States or PCT international application in the manner provided by the first paragraph of Title 35, United States Code § 112, I acknowledge the duty to disclose information which is material to patentability as defined in Title 37, Code of Federal Regulations § 1.56 which became available between the filing date of the prior application and the national or PCT international filing date of this application.

U.S. Parent Application Number	PCT Parent Number	Parent Filing Date (MM/DD/YYYY)	Parent Patent Number (*if applicable*)

[ ] Additional U.S. or PCT international application numbers are listed on a supplemental priority sheet attached hereto.

As a named inventor, I hereby appoint the following attorney(s) and/or agent(s) to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith:

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Brian M. Hoffman	**39,713**	**Michael W. Farn**	**41,015**

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I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Name of Sole or First Inventor: [ ] A petition has been filed for this unsigned inventor

Given Name	**James**	Middle Initial	**H.**	Family Name	**Parry**	Suffix e.g. Jr.	
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Residence: City	**Los Gatos**	State	**CA**	Country	**United States**	Citizenship	**U.S.**
Mailing Address	**P. O. Box 557**						
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[X] Additional inventors are being named on supplemental sheet(s) attached hereto